REMARKS

In accordance with the foregoing, claims have been neither amended nor canceled. Claims 1-15 pending and under consideration. No new matter has been presented.

ENTRY OF RESPONSE UNDER 37 C.F.R. §1.116

Applicant(s) request(s) entry of this Rule 116 Response and Request for Reconsideration because:

(a) the reference(s) applied to the claims are newly cited in the final Office Action, and Applicant(s) should be provided the opportunity to present patentability arguments and amendments in view thereof.

The Manual of Patent Examining Procedures sets forth in §714.12 that "[a]ny amendment that would place the case either in condition for allowance or in better form for appeal may be entered." (Underlining added for emphasis) Moreover, §714.13 sets forth that "[t]he Proposed Amendment should be given sufficient consideration to determine whether the claims are in condition for allowance and/or whether the issues on appeal are simplified." The Manual of Patent Examining Procedures further articulates that the reason for any non-entry should be explained expressly in the Advisory Action.

REJECTIONS UNDER 35 U.S.C. § 103

Claims 1, 4, 6, 9, and 11-14 are rejected under 35 U.S.C. § 103(a) as being unpatentable over Gao (U.S. Patent No. 6,449,590) in view of Ozawa (U.S. Patent No. 5,487,128) and further in view of Laflamme et al. ("16 Kbps Wideband Speech Coding Technique Based on Algebraic CELP," 1991).

Claim 1 recites "a speech characteristic classification unit, which stipulates a characteristic of speech corresponding to a current frame statistically using an open-circuit pitch value and a linear prediction coefficient in which a wide-band speech signal to be coded is perceptual weight filtered."

Gao discusses "[m]ore specifically, LP analysis at the block 239 is performed twice per frame but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized using predictive multi-stage quantization (PMVQ). The speech frame is divided into subframes. Parameters from the adaptive and fixed codebooks 257 and 261 are transmitted every subframe. The quantized and unquantized LP parameters or their interpolated versions are used depending on the subframe. An open-loop pitch lag is estimated at the block 241 once or twice per frame for PP mode or LTP mode, respectively."(col. 8, line 63 to col. 9, line 6).

Gao does not disclose a speech characteristic classification unit as recited in claim 1.

Further, Gao discusses "[a] voice/unvoiced classification and mode decision within the block 279 using the input speech s(n) and the residual $r_w(n)$ is derived where:"(col.12, lines 12-15).

As such, Gao does not disclose the invention as recited in claim 1. Neither does Ozawa or Laflamme et al.

Claim 1 recites "an adaptive codebook retrieving unit, which retrieves a pitch delay value around the open-circuit pitch value, calculates a pitch gain value, generates an adaptive codebook contribution signal corresponding to the retrieved pitch delay value, and outputs a difference between the generated adaptive codebook contribution signal and the perceptual weight filtered signal as a first fixed codebook target signal."

Gao discusses "[a]n open-loop pitch lag is estimated at the block 241 once or twice per frame for PP mode or LTP mode, respectively. Each subframe, at least the following operations are repeated. First, the encoder processing circuitry (operating pursuant to software instruction) computes x(n), the first target signal 229, by filtering the LP residual through the weighted synthesis filter W(z)H(z) with the initial states of the filters having been updated by filtering the error between LP residual and excitation."(col. 9, lines 4-12-emphasis added).

However, it is unclear how the examiner alleges that this addresses adaptive codebook searching.

Gao further discloses "[a]daptive codebook searching is performed on a subframe basis. It consists of performing closed-loop pitch lag search, and then computing the adaptive code vector by interpolating the past excitation at the selected fractional pitch lag. The LTP parameters (or the adaptive codebook parameters) are the pitch lag (or the delay) and gain of the pitch filter. In the search stage, the excitation is extended by the LP residual to simplify the close-loop search."(col. 21, lines 51-58).

As noted above, Gao discusses "closed-loop pitch lag search for searching an adaptive codebook."

Accordingly, Gao discusses an adaptive codebook retrieve unit but it is different from the features recited in claim 1.

Further, Ozawa merely discusses a first codebook and a second codebook and not the feature of claim 1 discussed above.

In addition, Laflamme et al. discusses "however, few studies have attempted to apply CELP to the context of wideband speech. The main drawback of CELP is its gross computational complexity. As the sampling frequency is doubled, larger frame sizes are needed to maintain a low bit rate transmission. Consequently, the user of mach larger excitation codebooks becomes

inevitable. For instance if we assume the same proportion bit rates and block lengths, the typical codebook size increases from a thousand entries (10bits) to a million entries (20 bits). Searching and string such a codebook size is rather impractical, unless some suboptimal approaches are utilized such multistage codebooks, or a split-band approach. From the above, discussion, it seems that it is impossible to use a full band approach for CELP coding of sideband speech.(see chapter 1 of Laflamme et al.-emphasis added).

It is noted that "it seems that it is impossible to use a full band approach for CELP coding of sideband speech" as disclosed in Laflamme et al.

As such, it is not only arguments being made by applicants but it is disclosed in Laflamme et al. That it is impossible to use a full hand approved.

Thus, Laflamme et al. clearly shows that teaching away from using a full band approach for CELP coding.

Accordingly, it is further respectfully submitted that the combination of Gao, Ozawa, and Laflamme et al. does not disclose or suggest that the features as recited in claim 1 for the above-discussed reasons.

Claim 4 recites "the second fixed codebook gain values include all gain values of each of the second fixed codebooks"

Gao discusses "[t]he codebook search for 4.55, 5.8, 6.65 and 8.0 kbps encoding bit rates consists of two steps. In the first step, a binary search of a single entry table representing the quantized prediction error is performed. In the second step, the index Index_1 of the optimum entry that is closest to the unquantized prediction error in mean square error sense is used to limit the search of the two-dimensional VQ table representing the adaptive codebook gain and the prediction error. Taking advantage of the particular arrangement and ordering of the VQ table, a fast search using few candidates around the entry pointed by Index_1 is performed. In fact, only about half of the VQ table entries are tested to lead to the optimum entry with Index_2. Only Index_2 is transmitted."(col. 35, lines 20-34).

As such, it is unclear whether Gao discloses the features recited in claim 4.

In addition, claim 4 is patentable due at least to its depending from claim 1, as well as for the additional recitations therein.

In addition, claims 6 and 9 are patentable due at least to the same or similar rationale as claims 1 and 4, respectively, as well as for the additional recitations therein.

Claims 11 and 12 are patentable due at least to the similar rationale as claim 1, as well as for the additional recitations therein.

Claims 2-3, and 7-8 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gao (U.S. Patent No. 6,449,590) in view of Ozawa (U.S. Patent No. 5,487,128) and further in view of Laflamme et al. ("16Kbps Wideband Speech Coding Technique Based on Algebraic CELP," 1991) and yet further in view of Chhatwal et al.(U.S. Patent No. 5,457, 783).

Claim 2 recites "wherein the second fixed codebook is composed of an algebraic codebook and a random codebook, and the second fixed codebook retrieving unit retrieves the random codebook in fricative sound or affricate section and retrieves the algebraic codebook in other sections."

However, Chhatwal et al. discusses "One solution to this problem would be to use a traditional random codebook based on noise-like waveforms in parallel with the bi-pulse codebook so that the bi-pulse codebook was used when it modeled the signal best, while the random codebook was used to model the certain types of unvoiced speech for which it was most appropriate. However, the disadvantage of this approach is that, as mentioned before, the random codebook is much more difficult to search than the bi-pulse codebook. The ideal solution would be to take the bi-pulse codebook vectors and transform them in some way such that they produced noise-like waveforms. Such an operation has the additional constraint that the transformation be easy to compute since this computation will be done many times in each frame."(col. 13, lines 50-63, emphasis added).

It is respectfully submitted that the combination of Gao, Ozawa, Laflamme et al., and Chhatwal et al., does not teach or suggest the invention as recited in claim 2, since Chhatwal et al. teaches away to use both bi-pulse codebook and random codebook.

In addition, claim 3 is patentable due at least to similar rationale as claim 1, as well as for the additional recitations therein.

Claims 7 and 8 are patentable due at least to similar rationale as claims 2 and 3, respectively, as well as for the additional recitations therein.

Claims 5 and 10 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gao in view of Ozawa and further in view of Laflamme et al. and yet further in view of Westerlund et al.(U.S. Patent No. 6,757,654).

Claim 5 recites "wherein the second fixed codebook gain values include a second standardized fixed codebook gain value and a ratio of the second standardized fixed codebook gain value and gain values of other second fixed codebooks."

Westerlund et al. discusses "the predictor could be updated based on energy changes present between frames. The encoder module could measure the distribution (e.g., ratio) between the LTP gain and the algebraic gain and send it with very few bit."(col. 21, lines 35-38).

As noted above, Westerlund et al. discusses the distribution between the LTP gain and the algebraic gain.

However, Weserlund et al. fails to disclose "wherein the second fixed codebook gain values include a second standardized fixed codebook gain value and a ratio of the second standardized fixed codebook gain value and gain values of other second fixed codebooks."

Thus, it is respectfully submitted that that the combination of Gao, Ozawa, Laflamme et al., and Westerlund et al. does not teach or suggest the invention as recited in claim 5.

Claim 10 is patentable due at least to the similar or the same rationale as claim 5, as well as for the additional recitations therein.

There being no further outstanding objections or rejections, it is submitted that the application is in condition for allowance. An early action to that effect is courteously solicited.

Finally, if there are any formal matters remaining after this response, the Examiner is requested to telephone the undersigned to attend to these matters.

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CONCLUSION

If there are any formal matters remaining after this response, the Examiner is requested to telephone the undersigned to attend to these matters.

If there are any additional fees associated with filing of this Amendment, please charge the same to our Deposit Account No. 19-3935.

Respectfully submitted,

STAAS & HALSEY LLP

Date: Apr. 7, 2008

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